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Avaya Solution & Interoperability Test Lab

Application Notes for Dolby Laboratories VCP9000 with Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.3 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Dolby Laboratories VCP9000 to interoperate with Avaya Aura® Session Manager R6.3 and Avaya Aura® Communication Manager R6.3. The Dolby Laboratories VCP9000 is a SIP conference phone that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and conferencing requirements.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Dolby Labs VCP9000 to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager Release 6.3. The Dolby Laboratories VCP9000 (VCP9000) is a SIP conference station that can register with Avaya Aura® Session Manager as a SIP endpoint combining the functionality of an IP phone and a conferencing station in support of voice communications and conferencing requirements.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the VCP9000 and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711, G.722, iLBC and G.729)
- Inbound calls
- Outbound calls
- Hold/Resume, Call Transfer and Conferencing
- Call termination (origination/destination)
- Avaya Features using FAC
    - Call Park
    - Call Pickup
    - Call Forward (Unconditional, Busy/no answer)
    - Find Me
- Message Waiting Indicator (MWI)
- Voicemail
- Serviceability

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of interoperability compliance testing was primarily on verifying call establishment on the VCP9000. The VCP9000 operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Facility Access Codes, and its interactions with Session Manager, Communication Manager, and other Avaya SIP, and H.323 phones were verified. The serviceability testing introduced failure scenarios to see if VCP9000 can recover from failures.

2.2. Test Results

The test objectives were verified. For serviceability testing, VCP9000 operated properly after recovering from failures such as network disconnects, and resets of VCP9000.

The features mentioned in **Section 2** worked successfully during compliance testing with the following exceptions, as these features are currently not supported by the VCP9000:

- Attended Call Transfer
- Blind Conference Call
- Long Hold Recall Timer
- Message Waiting Indication (MWI) – Since this phone is a conference room phone, the MWI feature would not be required.
- G729 Codec
- iLBC Codec is supported only between the VCP9000 endpoints
- At least one hardware-supported codec needs to be listed on VCP9000 for iLBC or G.722 to work.

2.3. Support

For technical support on Dolby Labs VCP9000, please contact via the following:

- Web: http://www.dolby.com

3. Reference Configuration

Once VCP9000 registers as a SIP endpoint with Session Manager, it can place and receive voice calls with various supported features as listed above in **Section 2.1**. The reference configuration used for the compliance test is shown in **Figure 1** below.

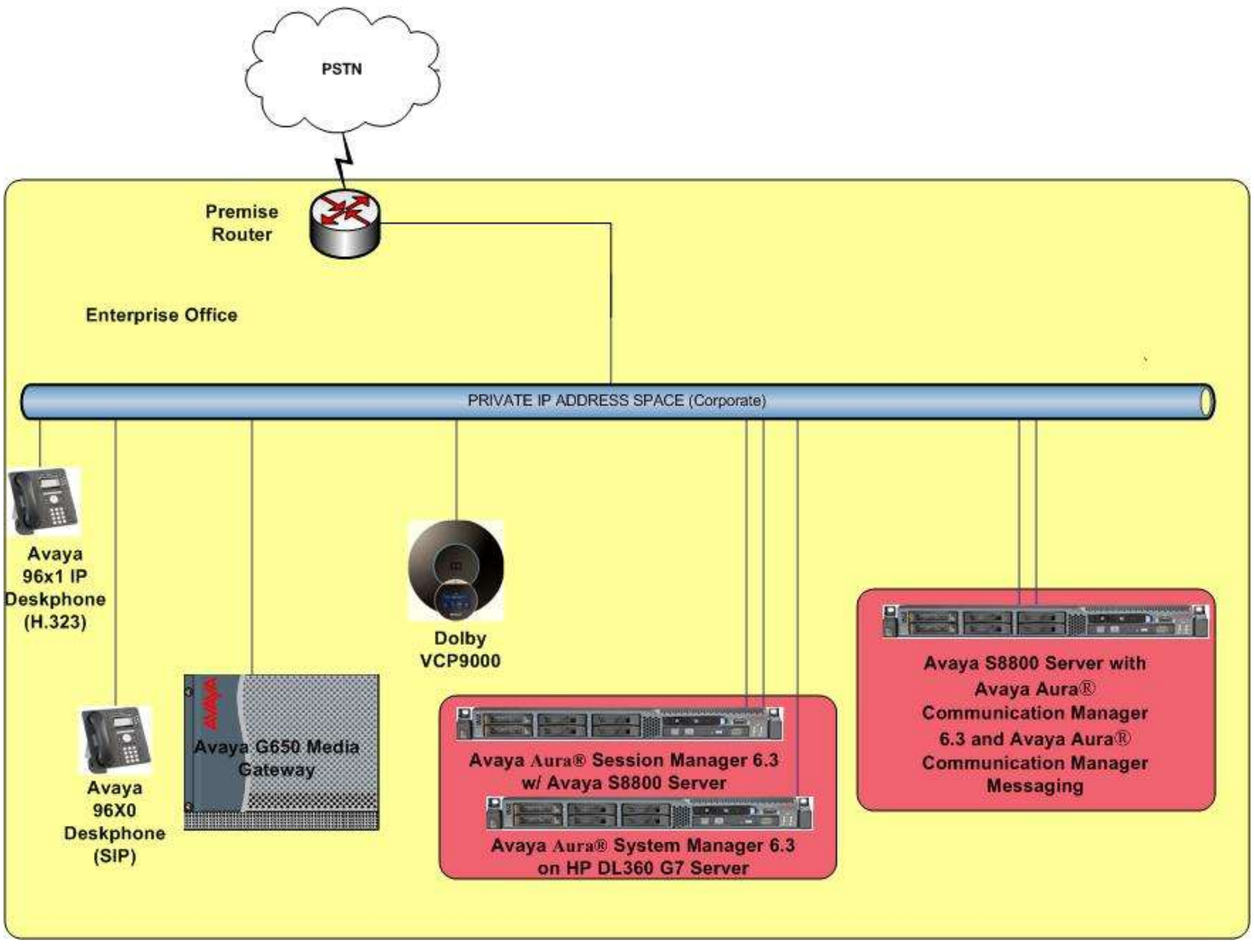


Figure 1: Dolby Labs VCP9000 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya Aura® System Manager running on an Avaya HP DL360/G7 Server	R6.3.12.9.3022 – SP12
Avaya Aura® Session Manager running on an Avaya S8800 Server	R6.3.12.0.631208 – SP12
Avaya Aura® Communication Manager running on Avaya S8800 Server and G450 Media Gateway	R016x.03.0.124.0 – R6.3, SP10
Avaya 96x1 IP Deskphone (H323)	R6.2.2313
Avaya 96x0 IP Deskphone (SIP)	R2.6.9.1
Dolby Labs VCP9000	2.1.0.24

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- SIP Entities and corresponding Entity Links for Session Manager and Communication Manager.
- Define Communication Manager as Administrable Entity (i.e., Managed Element).
- Application Sequence.
- Add SIP Users.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "**https://<*ip-address*>/SMGR**", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice.

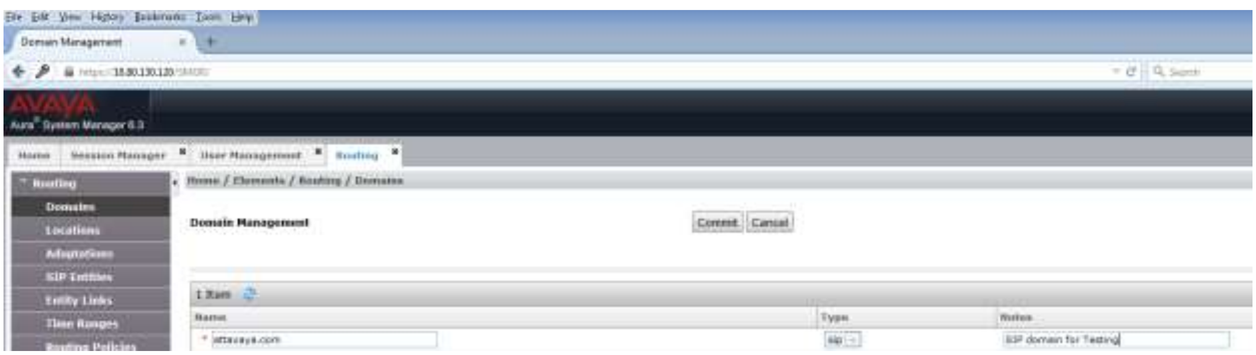
5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. It can be done by selecting **Domains** on the left and clicking the **New** (not shown) button on the right.

The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *attavaya.com*).
- **Notes:** Descriptive text (optional).

Click **Commit**.



5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

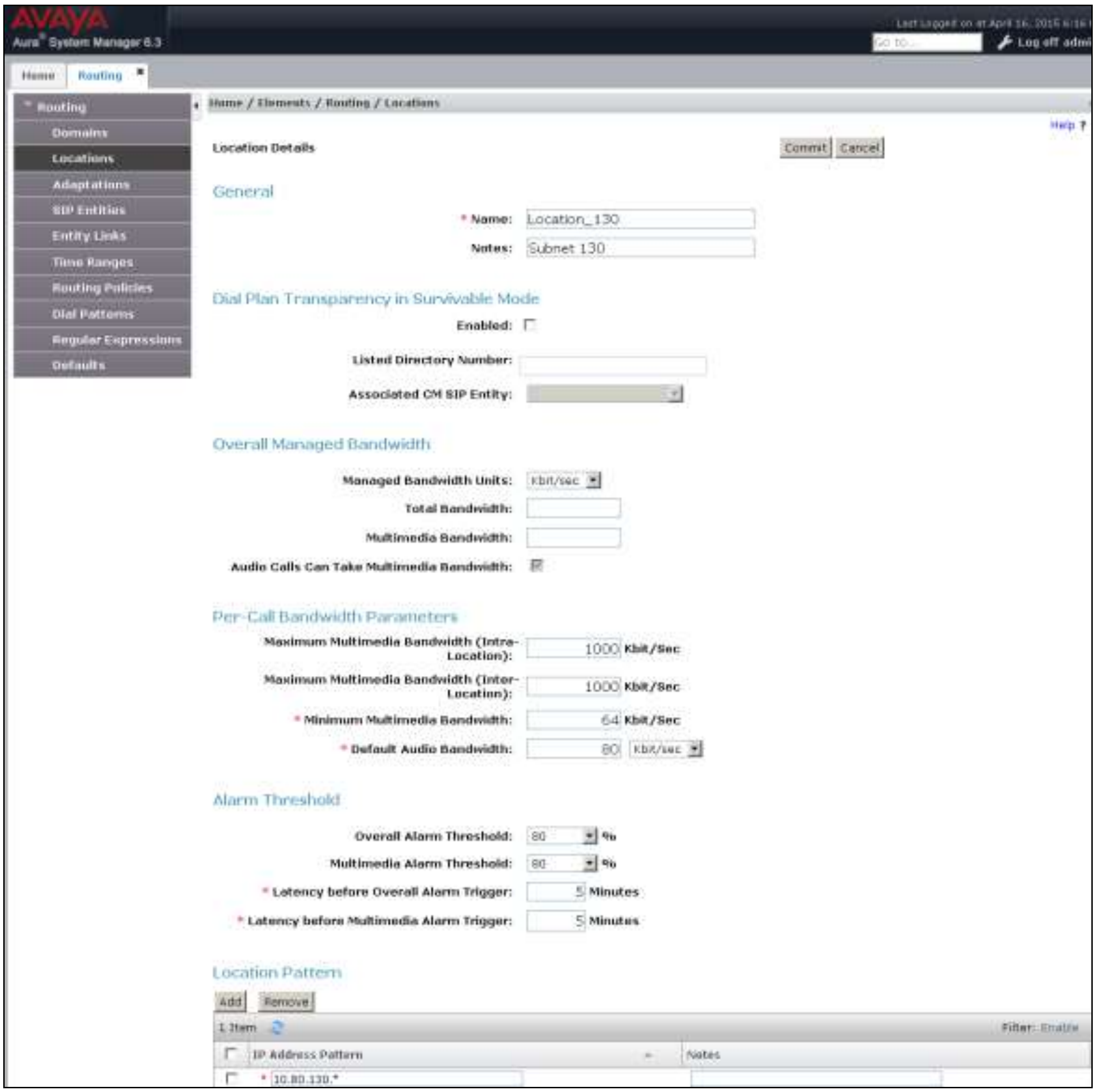
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *Location_130* location for Communication Manager and similarly a location was defined for Session Manager. Click **Commit** to save the Location definition.



5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and Communication Manager. The screens below also show the corresponding Entity Links.

5.3.1. Session Manager Entity

To add a SIP Entity, navigate to **Home→Elements→Routing→SIP Entities**, and click on the **New** (not shown and configure as follows:

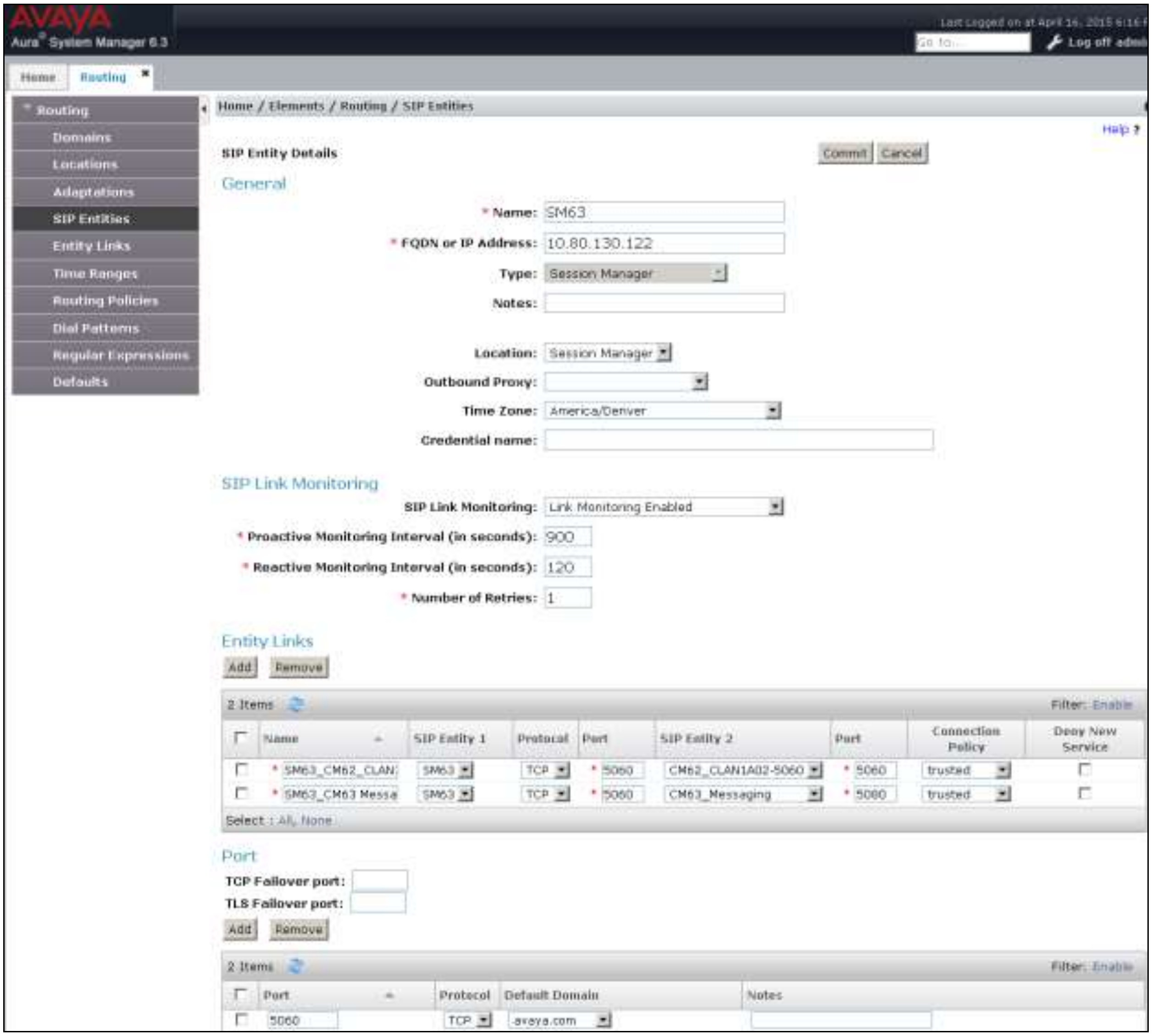
Under ***General***:

- **Name:** Any descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Under ***Port***, click **Add**, and then edit the fields in the resulting new row as shown below:

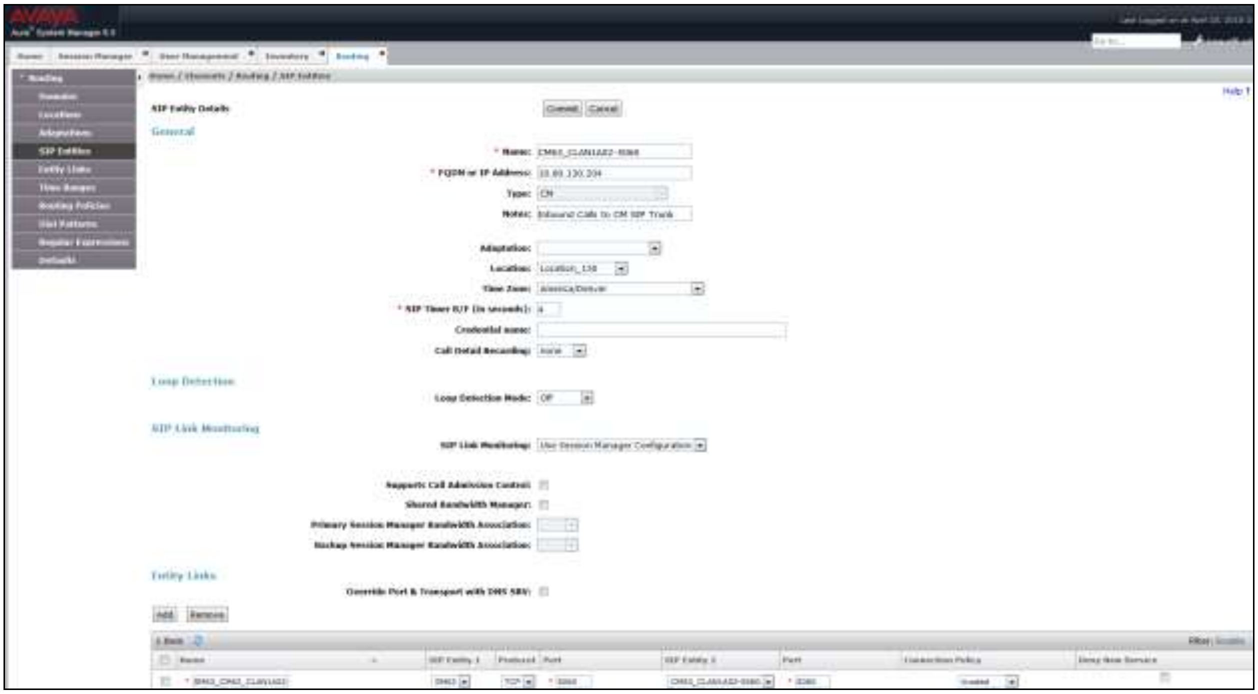
- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise (e.g. *attavaya.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.


AVAYA
Aura® System Manager 6.3
Home / Elements / Routing / SIP Entities
SIP Entity Details
General
* Name: SM63
* FQDN or IP Address: 10.80.130.122
Type: Session Manager
Notes:
Location: Session Manager
Outbound Proxy:
Time Zone: America/Denver
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds): 900
* Reactive Monitoring Interval (in seconds): 120
* Number of Retries: 1
Entity Links
Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy | Deny New Service
SM63_CM62_CLAN | SM63 | TCP | 5060 | CM62_CLAN1A02-5060 | 5060 | trusted
SM63_CM63 Messa | SM63 | TCP | 5060 | CM63_Messaging | 5060 | trusted
Port
TCP Failover port:
TLS Failover port:
Port | Protocol | Default Domain | Notes
5060 | TCP | avaya.com

5.3.2. Communication Manager Entity

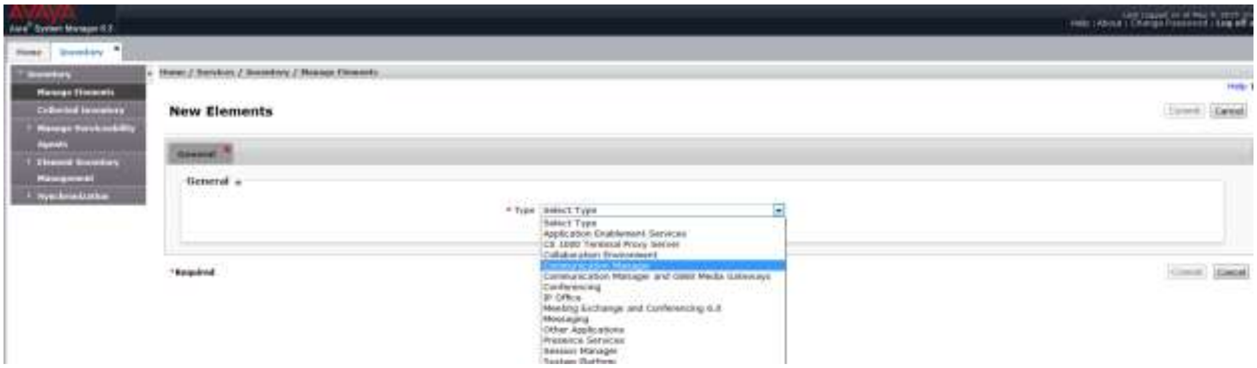
The following screen displays the Communication Manager entity configured for this reference configuration.



5.4. Define Communication Manager as a Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

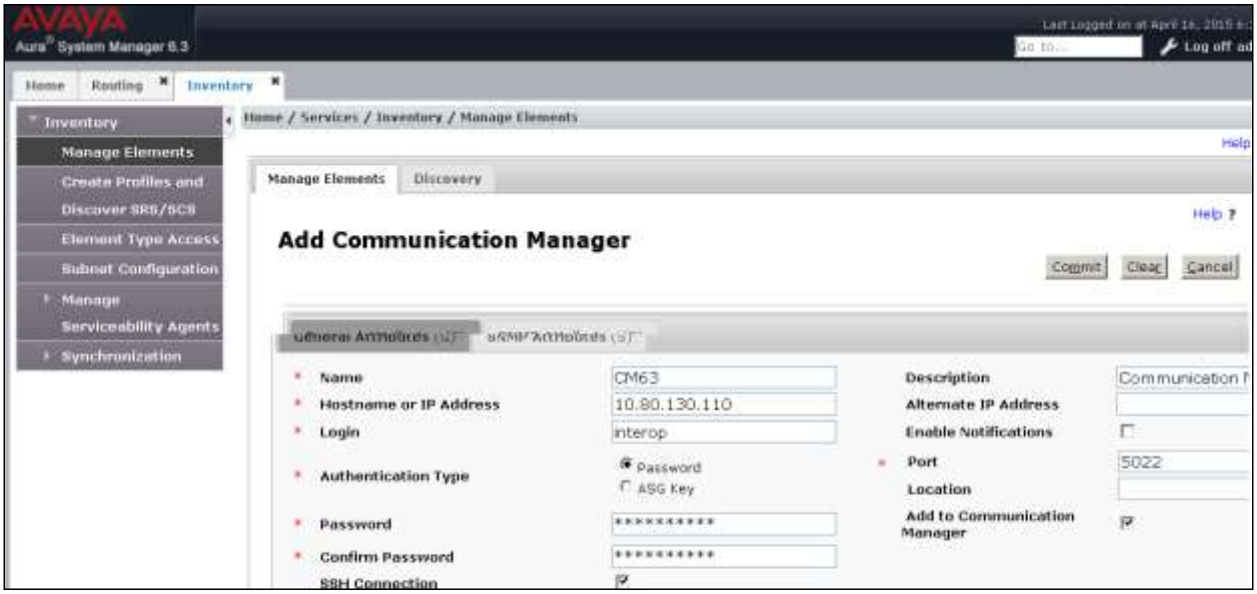
To define Communication Manager as a managed element, navigate to **Home→Services→ Inventory→Manage Elements** on the left and click on the **New** (not shown) button on the right. In the **Type** field that is displayed, select *Communication Manager*.



In the **Add Communication Manager** screen, fill in the following fields as follows:
Under *General Attributes*:

- **Name:** Enter an identifier for Communication Manager.
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager.
- **Login:** Enter the login used for administration access to Communication Manager.
- **Authentication Type:** Select the **Password** button.
- **Password** Enter a valid password.
- **Confirm Password** This should match the password entered in the **Password** field above.

Click **Commit** to save.



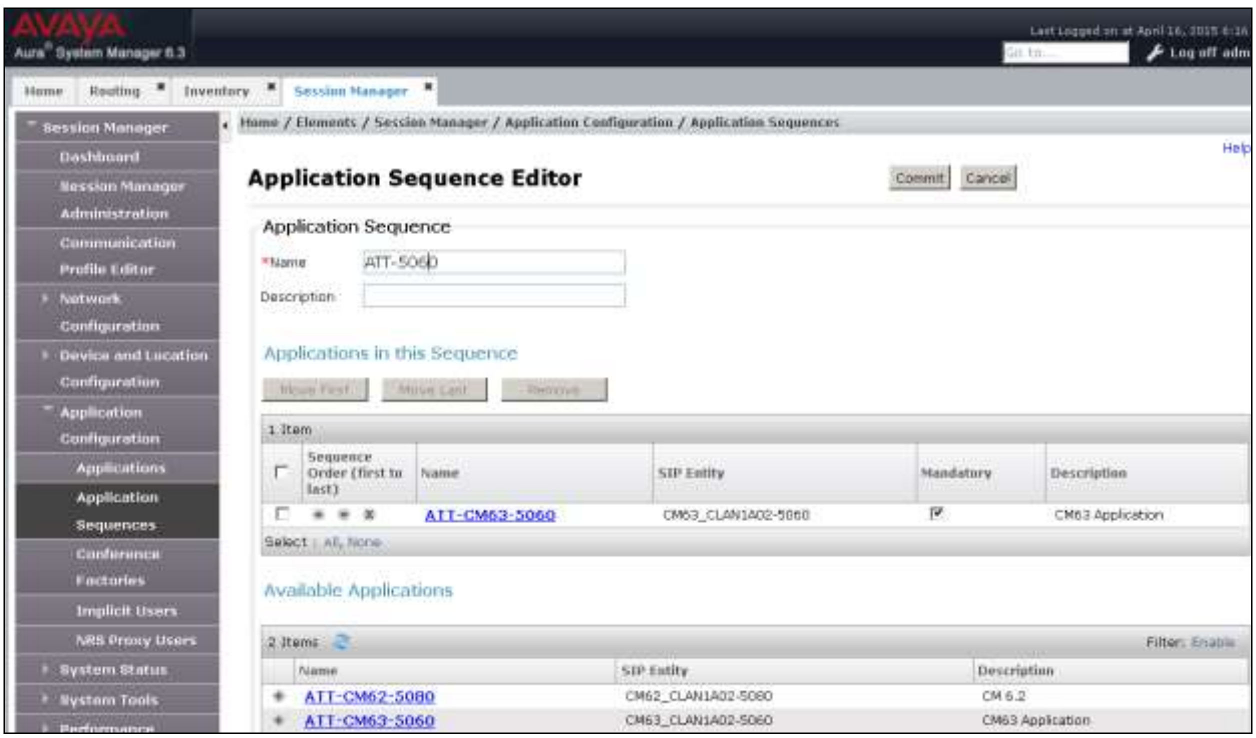
5.5. Add Application Sequence

Navigate to **Home→Elements→Session Manager→Application Configuration→Applications** and configure as follows:

- **Name:** Enter any descriptive name.
- **SIP Entity:** Select the Communication Manager SIP Entity configured in **Section 5.3.2**.
- **CM System for SIP Entity:** Select the system configured in **Section 5.4**.

Click **Commit** to save the Application definition.

Next, define the **Application Sequence** for Communication Manager as shown below.

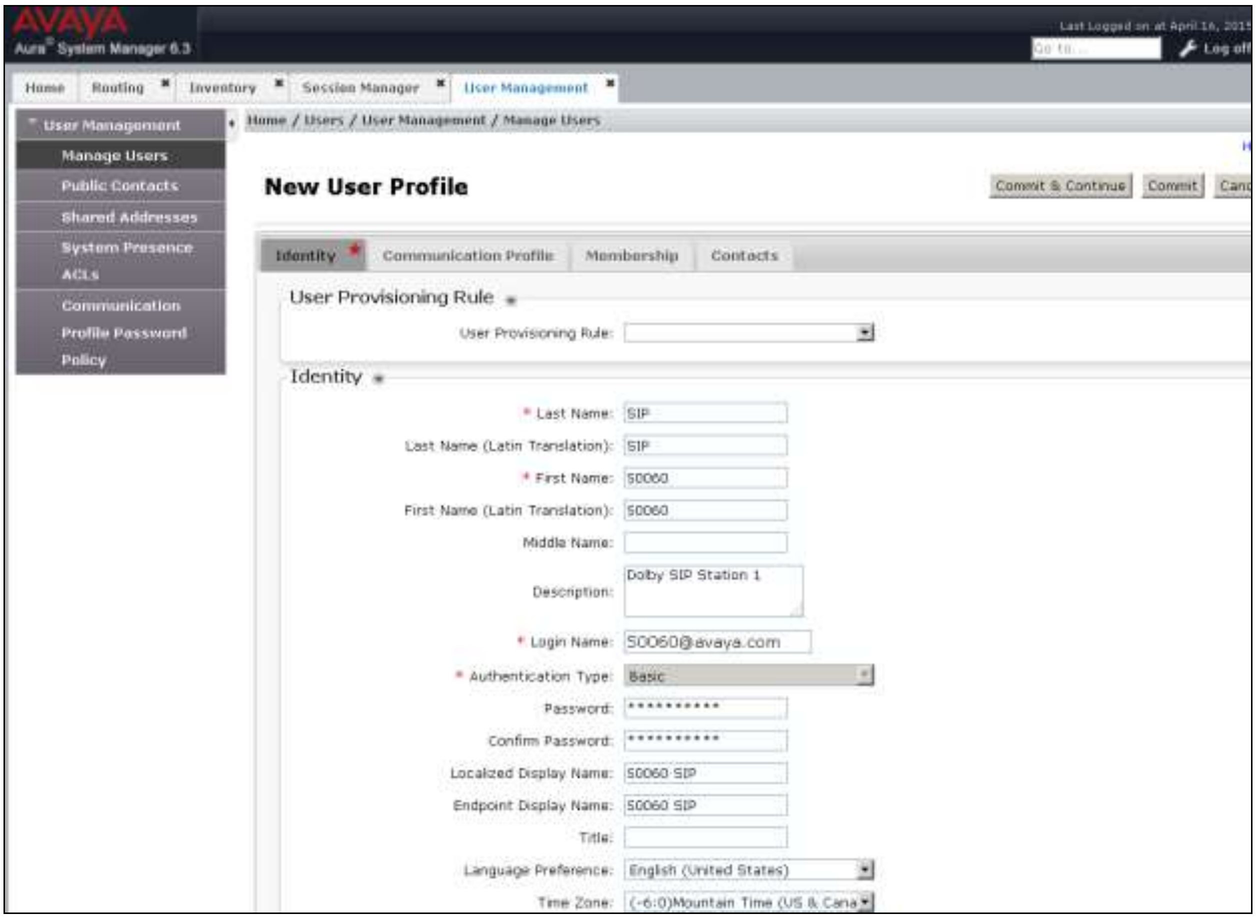


5.6. Add SIP Users

VCP9000 was entered as a SIP user on Session Manager using the following steps. This configuration is automatically synchronized with Communication Manger as verified in **Section 6.3**.

Enter values for the following required attributes for a new SIP user in the new user form:

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., *50060@attavaya.com*).
- **Password:** Enter the password which will be used to log into System Manager.
- **Confirm Password:** Re-enter the password from above.



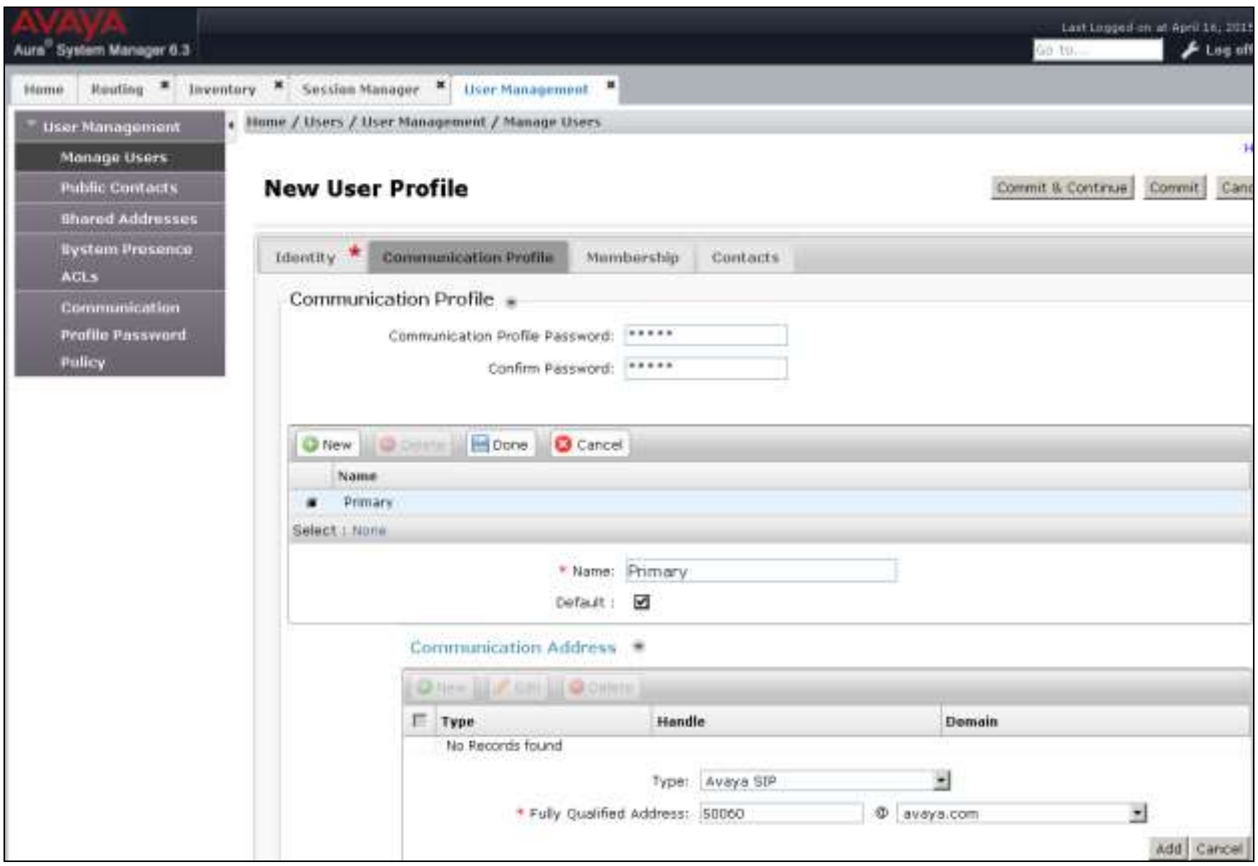
Click the **Communication Profile** tab and select **New** (not shown) to define a **Communication Profile** for a new SIP user. Enter values for the following required fields:

- **Communication Profile Password:** Enter a valid password.
- **Confirm Password:** Make sure that it matches the password entered above.
- **Name:** Enter name of the communication profile.
- **Default:** Check box to indicate that it is the default profile.

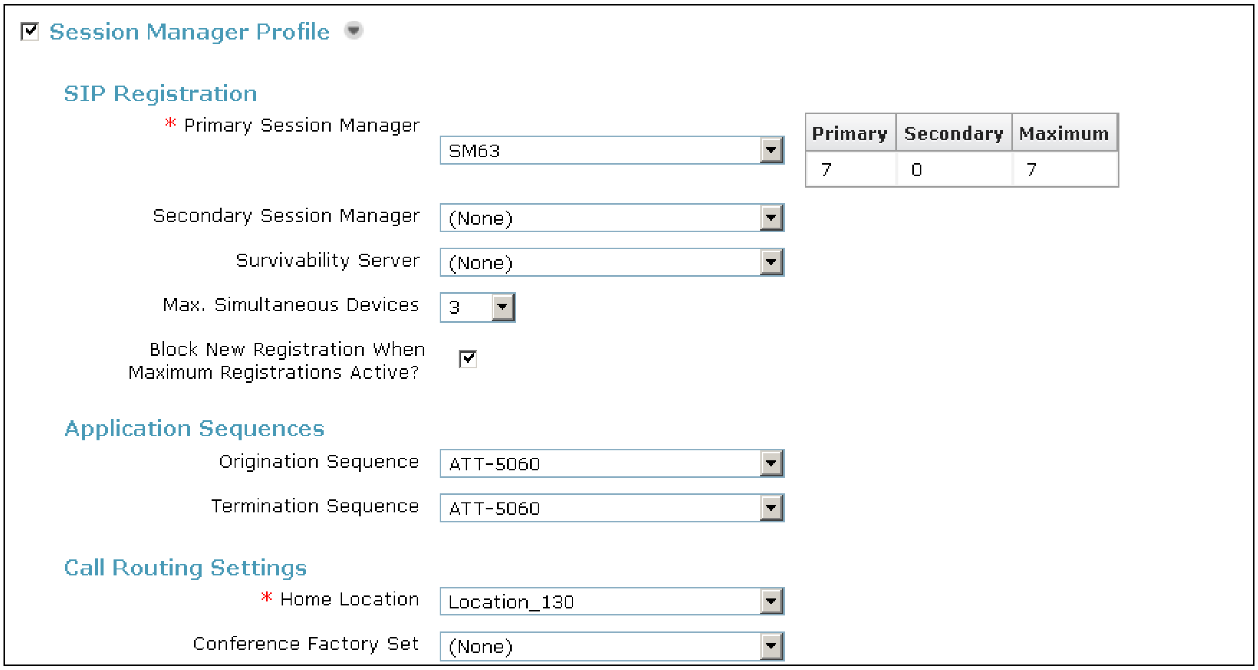
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

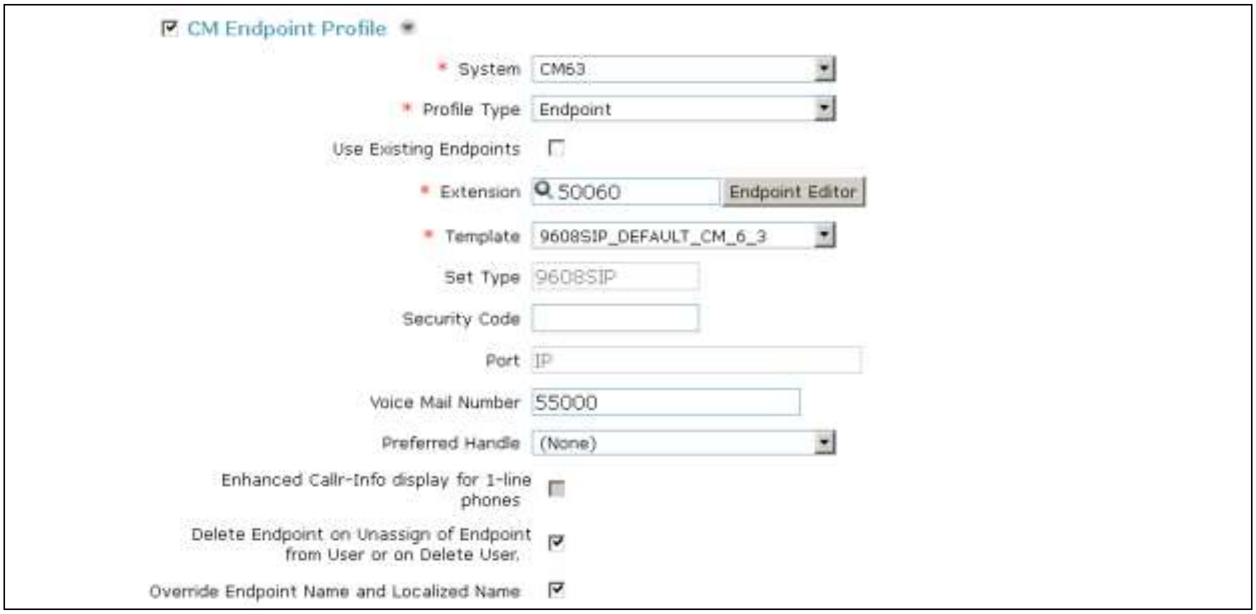


In the **Session Manager Profile** section, specify the Session Manager entity configured in **Section 5.3.1** and assign the **Application Sequence** defined in **Section 5.5** to both the **Originating Sequence** and **Termination Sequence** fields. Additionally, set **Home Location** field to **Location_130** configured in **Section 5.2**.



In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager in **Section 5.4**
- **Profile Type:** Select *Endpoint.*
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone which is set to **9608SIP_DEFAULT_CM_6_3** for **VCP9000**



6. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the VCP9000 as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

6.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                                 Software Package: Enterprise
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 6400  25
                                      Maximum Stations: 2400  10
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  0
                    Maximum Off-PBX Telephones -   OPS: 9600  5
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  15
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

6.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for Session Manager (ASM63) and Communication Manager (CLAN_1A02). The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
    Name              IP Address
default             0.0.0.0
ASM63               10.80.130.122
CLAN_1A02           10.80.130.204
procr               10.80.130.110
procr6              ::

( 4 of 4   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *attavaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '2) is specified in the SIP signaling group.

```
change ip-network-region 2                                          Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: attavaya.com
    Name: Main Network Region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                         RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the VCP9000. The form is accessed via the **change ip-codec-set 2** command. Note that IP codec set **2** was specified in IP Network Region '2 shown above. The following form shows the list of codecs tested. The order of these codecs was changed to support the some of the codecs for reasons listed in **Section 2.2**.

```
change ip-codec-set 2                                               Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.711A            n         2        20
 3: G.722-64K                   2        20
 4: iLBC                        1       20-30
 5:
 6:
 7:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the recommended TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *attavaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 2                                            Page   1 of   1
                                SIGNALING GROUP

Group Number: 2             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: CLAN_1A02            Far-end Node Name: ASM63
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 2
                                   Far-end Secondary Node Name:
Far-end Domain: attavaya.com
                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 2                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: SIP Endpoints/CM Messaging  COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 15
```

On **Page 3** of the **Trunk Group** form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 2                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
 DSN Term? n
```

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with "**5**" and whose calls are routed over any trunk group, including SIP trunk group "**2**", have the number sent to the far-end for display purposes.

```
change private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  33               10                          5     Total Administered: 4
 5  58               10                          5        Maximum Entries: 540
 5  5                2                           5
 5  600              10                          5
```

6.3. Verify SIP Stations

Use the **display station** command to view each VCP9000 SIP endpoint configured in **Section 5.6**.

```
display station 50060                                             Page   1 of   6
                                     STATION

Extension: 40012                          Lock Messages? n              BCC: 0
     Type: 9620SIP                        Security Code:                 TN: 1
     Port: S00003                       Coverage Path 1: 1              COR: 1
     Name: 50060 SIP                    Coverage Path 2:                COS: 1
                                        Hunt-to Station:
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19
                                                  Message Lamp Ext: 40012

        Display Language: english

          Survivable COR: internal
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n
```

Use the **display off-pbx-telephone station-mapping** to verify proper entry of VCP9000 SIP station in Communication Manager.

```
display off-pbx-telephone station-mapping 50060                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk        Config  Dual
 Extension                 Prefix                     Selection    Set     Mode
 50060            OPS          -        50060          aar         1
```

On **Page 2**, verify that the **Call Limit** matches the number of *call-appr* entries in the station form.

```
display off-pbx-telephone station-mapping 50060                   Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 50060         OPS     3           both       all         none
```

7. Configure Dolby Labs VCP9000

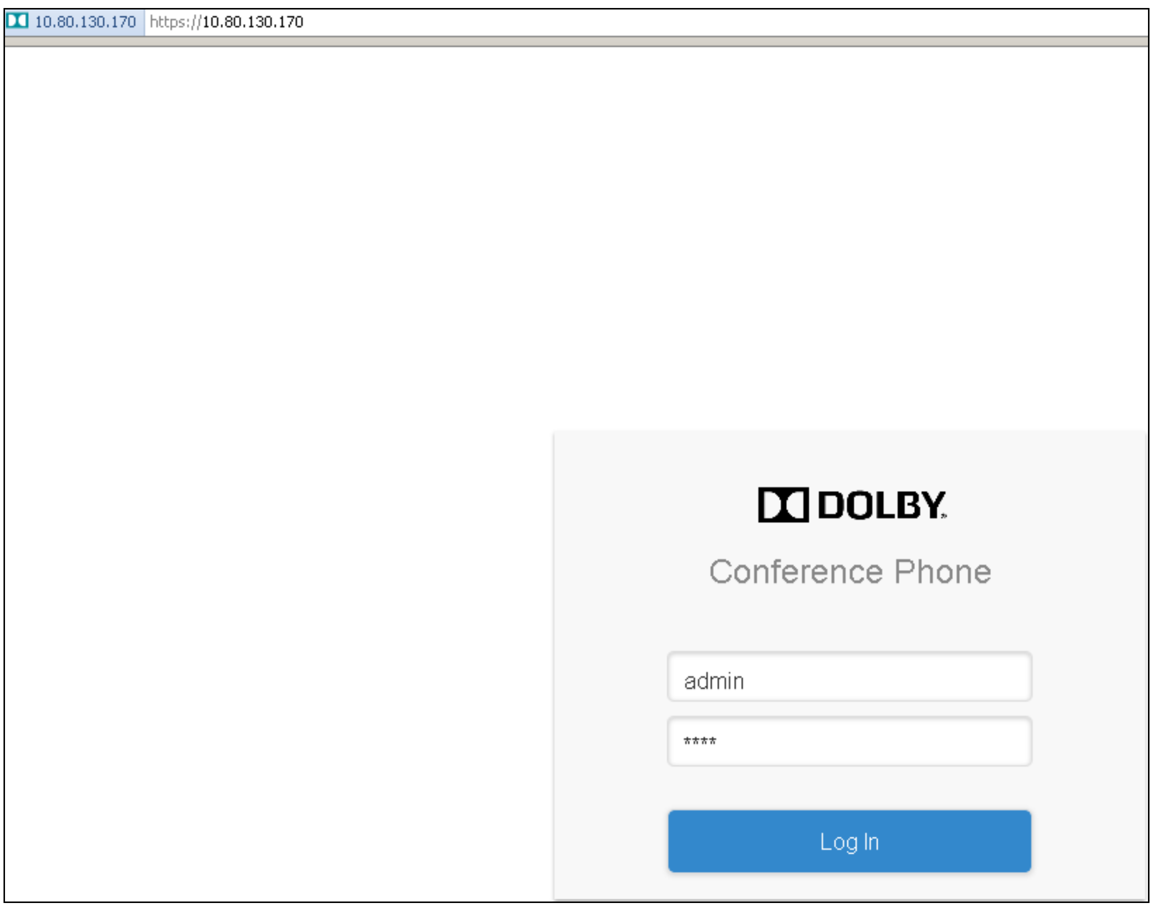
This section provides the procedures for configuring the VCP9000. The procedures include the following areas:

- Configure IP address on VCP9000
- Launch Web Interface
- Administer network settings
- Administer SIP settings
- Configure media settings

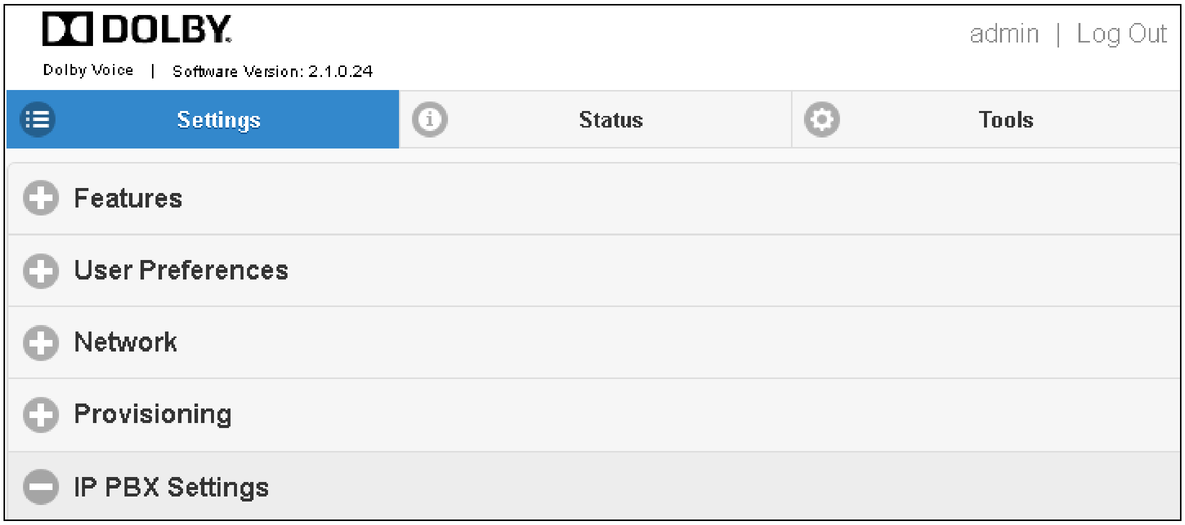
7.1. Launch Web Interface

In the test configuration, the IP address along with the gateway information was changed from VCP9000 and then rest of the configuration was done using the web access. To configure the IP address, click on the **Settings** icon. Administration privilege is required to configure IP address.

Access the VCP9000 web-based interface by using the URL "**https://<ip-address>**" in an Internet browser window, where <ip-address> is the IP address of the VCP9000 endpoint. The Dolby Labs homepage screen is displayed, as shown below. Log in using the appropriate credentials.



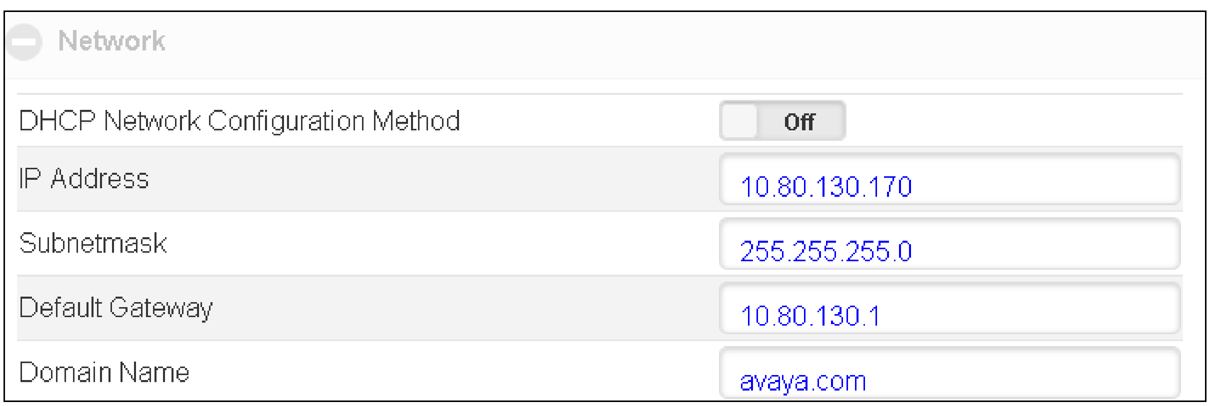
The following screen shows the landing page for VCP9000.



7.2. Verify/Modify Network Settings

To verify/modify networks settings, navigate to **Settings→Network** from the menu on the left and enter the following values for the specified fields. Retain default values in the remaining fields. While the default setting for **Connection Type** is DHCP, for the compliance test a static IP address was used. DHCP may be used if supported at the customer site.

- **DHCP Network Configuration Method:** Set to **Off** in this reference configuration.
- **IP Address:** Verify/Modify the IP address.
- **Subnetmask:** Verify/Modify the subnet mask.
- **Default Gateway:** Verify/Modify the default gateway.
- **Domain Name:** Verify/Modify the domain as configured in **Section 5.1**.

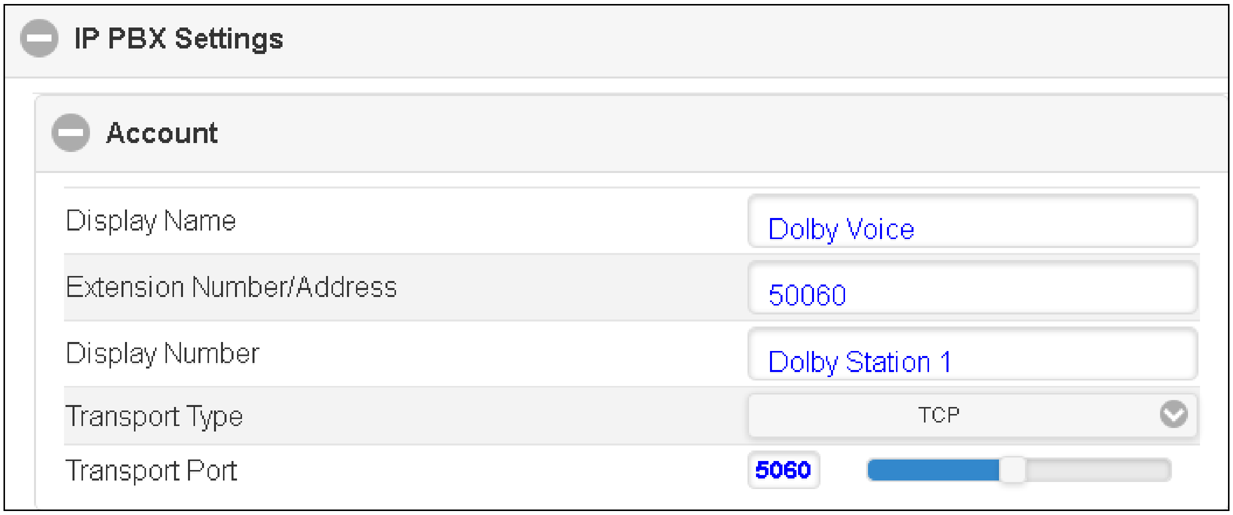


7.3. Administer SIP Settings

7.3.1. Administer SIP Account

To configure SIP Registration Settings, navigate to **Settings→IP PBX Settings→Account** from the menu on the left and enter the following values for the specified fields. Default values were used for the remaining fields.

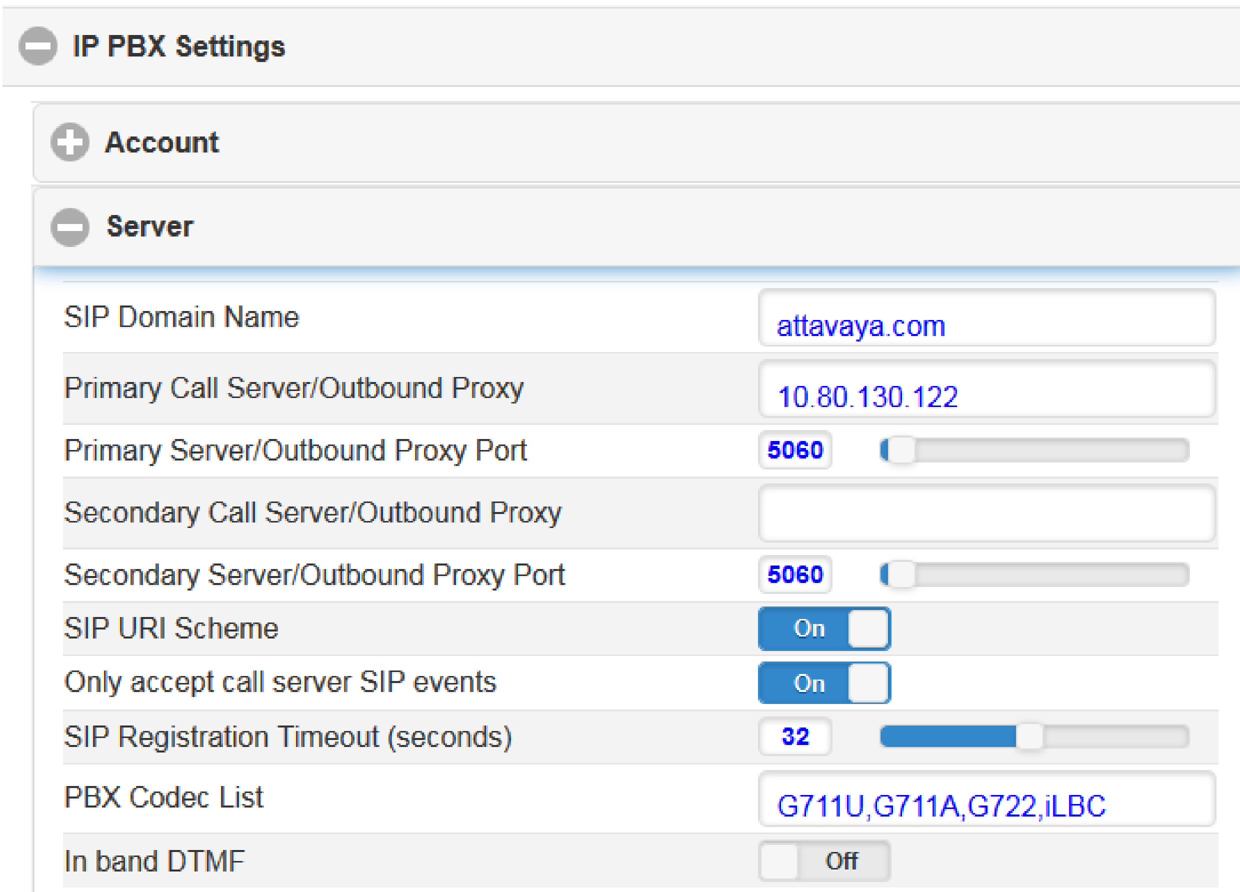
- **Display Name:** Enter any descriptive name
- **Extension Number/Address:** Enter the extension in Login Name from **Section 5.6**.
- **Display Number:** Enter any valid string.



7.3.2. Administer SIP Server

To administer the SIP Server settings, navigate to **Settings→IP PBX Settings→Server** from the menu on the left and enter the following values for the specified fields. Default values used for the remaining fields.

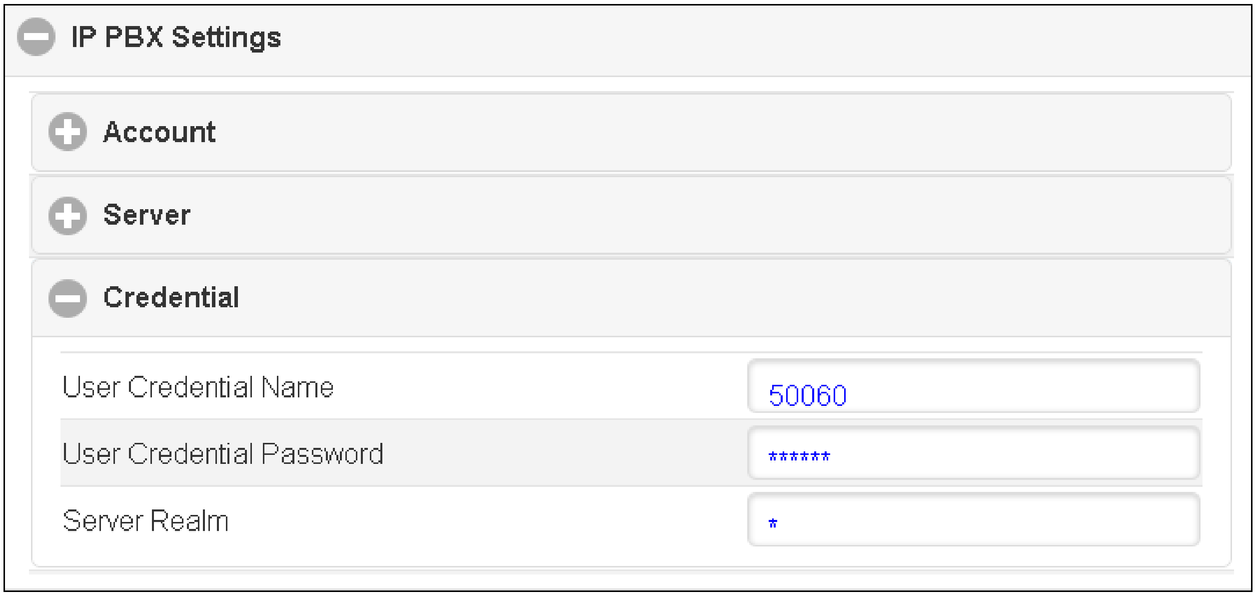
- **SIP Domain Name:** Set to domain name configured in **Section 5.1**.
- **Primary Call Server/Outbound Proxy:** Set to the IP address of the Session Manager Signaling interface from **Section 5.3.1**.
- **PBX Codec List:** The order of the codec list shown below can be changed.



7.3.3. Administer SIP Credentials

To administer the SIP Credential settings, navigate to **Settings→IP PBX Settings→Credential** from the menu on the left and enter the following values for the specified fields.

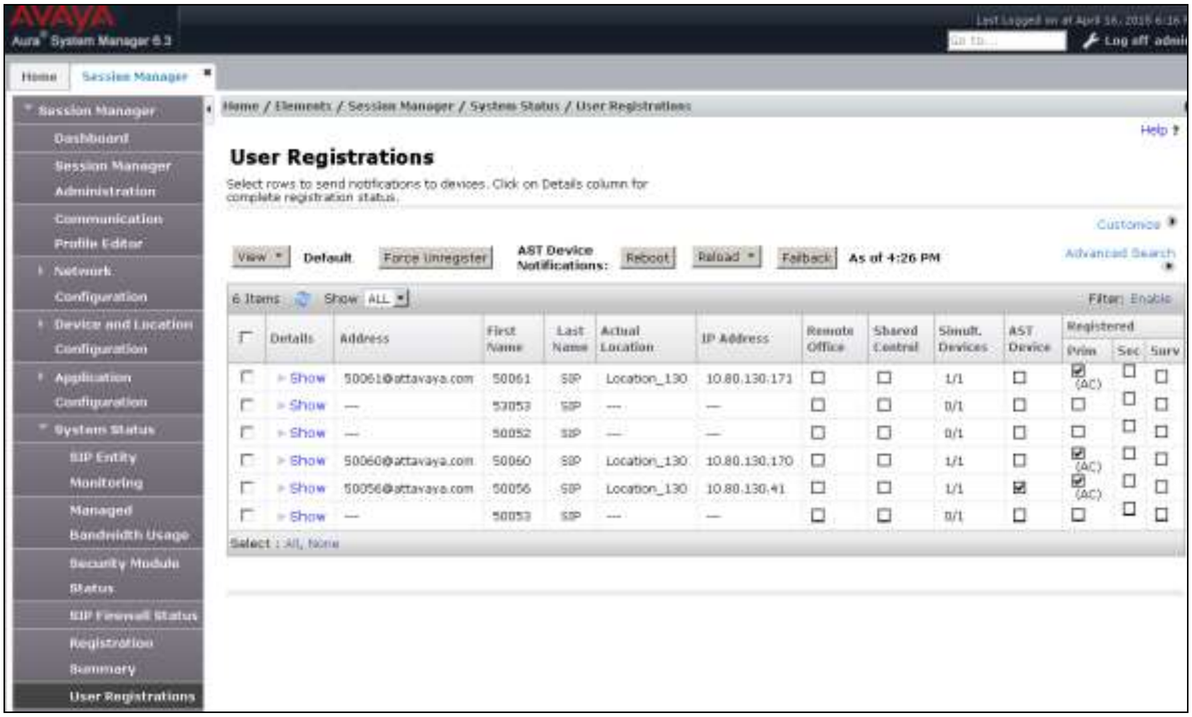
- **User Credential Name:** Set to the user part of **Login Name** configured in **Section 5.6**.
- **User Credential Password:** Set to the **Communication Profile** password defined in **Section 5.6**.
- **Server Realm:** Set to * to accept all realms.



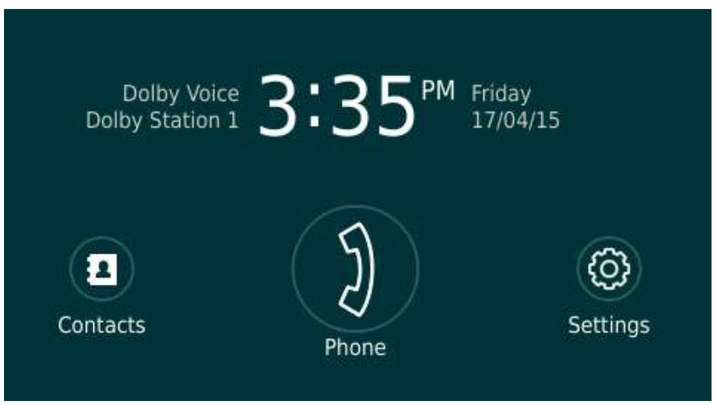
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Communication Manager with VCP9000.

- Verify that VCP9000 is registered with Session Manager. The following screen shows the registered SIP users with Session Manager:



- The following screen shows that VCP9000 is successfully registered with Session Manager. If the registration was not successful, the **Phone** button is not enabled and the **Settings** button on the phone has a red exclamation sign as shown here



- Verify that basic calls can be made from and to VCP9000 and another telephone registered with Communication Manager.

9. Conclusion

These Application Notes describe the configuration steps required for Dolby Labs VCP9000 SIP conference station to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed.

10. Additional References

This section references the product documentation available at support.avaya.com relevant to these Application Notes.

1. Administering Avaya Aura® Session Manager, Release 6.3
2. Maintaining and Troubleshooting Avaya Aura® Session Manager, Release 6.3
3. Administering Avaya Aura® System Manager, Release 6.3
4. Administering Avaya Aura® Communication Manager
5. *Dolby Labs VCP9000 SIP Conference Telephone User Guide* available at http://www.dolby.com.